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APPLICATION NO.	FILING DATE	FIRST NAMED INVENTOR	ATTORNEY DOCKET NO.	CONFIRMATION NO.
10/045,458	11/07/2001	Frank Baumgarte	Baumgarte 1-6-8	4306

46900 7590 05/02/2007
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EXAMINER	
TRAN, CON P	
ART UNIT	**PAPER NUMBER**
2615	
MAIL DATE	**DELIVERY MODE**
05/02/2007	PAPER

Please find below and/or attached an Office communication concerning this application or proceeding.

The time period for reply, if any, is set in the attached communication.

Office Action Summary	Application No.	Applicant(s)
	10/045,458	BAUMGARTE ET AL.
	Examiner	Art Unit
	Con P. Tran	2615

-- The MAILING DATE of this communication appears on the cover sheet with the correspondence address --

Period for Reply

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE 3 MONTH(S) OR THIRTY (30) DAYS, WHICHEVER IS LONGER, FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133). Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

Status

1)☒ Responsive to communication(s) filed on *09 February 2007*.

2a)☐ This action is **FINAL**. 2b)☒ This action is non-final.

3)☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

Disposition of Claims

4)☒ Claim(s) *1-77* is/are pending in the application.

4a) Of the above claim(s) _____ is/are withdrawn from consideration.

5)☐ Claim(s) _____ is/are allowed.

6)☒ Claim(s) *1-77* is/are rejected.

7)☐ Claim(s) _____ is/are objected to.

8)☐ Claim(s) _____ are subject to restriction and/or election requirement.

Application Papers

9)☐ The specification is objected to by the Examiner.

10)☐ The drawing(s) filed on _____ is/are: a)☐ accepted or b)☐ objected to by the Examiner.

Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).

Replacement drawing sheet(s) including the correction is required if the drawing(s) is objected to. See 37 CFR 1.121(d).

11)☐ The oath or declaration is objected to by the Examiner. Note the attached Office Action or form PTO-152.

Priority under 35 U.S.C. § 119

12)☐ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).

a)☐ All b)☐ Some * c)☐ None of:

1.☐ Certified copies of the priority documents have been received.

2.☐ Certified copies of the priority documents have been received in Application No. _____.

3.☐ Copies of the certified copies of the priority documents have been received in this National Stage application from the International Bureau (PCT Rule 17.2(a)).

* See the attached detailed Office action for a list of the certified copies not received.

Attachment(s)

1) ☒ Notice of References Cited (PTO-892)

2) ☐ Notice of Draftsperson's Patent Drawing Review (PTO-948)

3) ☒ Information Disclosure Statement(s) (PTO/SB/08) Paper No(s)/Mail Date *01/19/07*.

4) ☐ Interview Summary (PTO-413) Paper No(s)/Mail Date. _____ .

5) ☐ Notice of Informal Patent Application

6) ☐ Other: _____.

DETAILED ACTION

Continued Examination Under 37 CFR 1.114

1. A request for continued examination under 37 CFR 1.114, including the fee set forth in 37 CFR 1.17(e), was filed in this application after final rejection. Since this application is eligible for continued examination under 37 CFR 1.114, and the fee set forth in 37 CFR 1.17(e) has been timely paid, the finality of the previous Office action has been withdrawn pursuant to 37 CFR 1.114. Applicant's submission filed on February 09, 2007 has been entered.

Claim Objections

2. Claims 10, and 11 are objected to because of the following informalities: In both claims, line 1 read "The invention of claim 1, wherein step (b) comprises"; however, there is no "step (b)" in claim 1. For purpose of examining, Examiner assumes Applicant intends to claim in the embedding step.

Appropriate correction is required.

Claim Rejections - 35 USC § 102

3. The following is a quotation of the appropriate paragraphs of 35 U.S.C. 102 that form the basis for the rejections under this section made in this Office action:

A person shall be entitled to a patent unless –

(e) the invention was described in (1) an application for patent, published under section 122(b), by another filed in the United States before the invention by the applicant for patent or (2) a patent granted on an application for patent by another filed in the United States before the invention by the applicant for patent, except that an international application filed under the treaty defined in section 351(a) shall have the effects for purposes of this subsection of an application filed in the United States only if the international application designated the United States and was published under Article 21(2) of such treaty in the English language.

4. **Claims 27-38, 49, and 52 are** rejected under 35 U.S.C. 102(e) as being anticipated by Knappe et al. U.S. Patent 6,850,496 (hereinafter, "Knappe"), cited by Applicants.

Regarding **claim 27**, Knappe teaches a method for encoding C input audio channels (36L, 36R, Endpoint C, Figs. 3, 4, 5; col. 4, lines 46-51) to generate E transmitted audio channels (common transmit channel; col. 5, lines 46-49), the method (see Figs. 3, 4, 5, 6, and respective portions of the specification), comprising:

providing two or more of the C input channels (34L, 34R, Endpoint C, Figs. 3, 4, 5; col. 4, lines 46-51) in a frequency domain (subband coding, col. 3, lines 54-64);

generating one or more cue codes (ILD/ITD) for each of two or more different frequency bands in the two or more input channels in the frequency domain (subband coding, col. 3, lines 54-64; manipulate ILD for all frequencies, or just subbands containing higher frequencies; col. 8, lines 44-63); and

downmixing the C input channels to generate the E transmitted channels, where $C > E \geq 1$ (using common transmit channel for all capture channels at a given endpoint; col. 5, lines 46-49; Fig. 5, col. 6, lines 1-10) such that an audio decoder

(including decoder 86, Fig. 6) is enabled to generate more than E different playback audio channels based on only the E transmitted channels (i.e., translate monophonic to a traditional two-speaker format) and the one or more cue codes (ILD/ITD; col. 8, lines 1-14; see Fig. 6).

Knappe thus teaches all the claimed limitations.

Regarding **claim 28,** Knappe teaches the invention of claim 27, further comprising formatting the E transmitted channels and the one or more cue codes into a transmission format such that:

the format (monophonic, col. 8, lines 1-14) enables a first audio decoder (including decoder 86, Fig. 6; col. 6, lines 59-63) having no knowledge of the existence of the one or more cue codes to generate E playback audio channels (mono) based on the E transmitted channels and independent of the one or more cue codes (no further localization can be performed; col. 10, lines 8-12); and

the format enables (stereophonic) a second audio decoder (including decoder 84, Fig. 6; col. 6, lines 59-63) having knowledge of the existence of the one or more cue codes (ILD/ITD) to generate more than E playback audio channels (stereo; left , right) based on the E transmitted channels and the one or more cue codes (single channel mapped to two channels with appropriate ILD/ITD to each channel; col. 8, lines 1-14; see Fig. 6).

Regarding **claim 29,** Knappe teaches the invention of claim 28 wherein the format enables the second audio decoder (including decoder 84, Fig. 6; col. 6, lines 59-63) to generate C playback audio channels (two channels) based on the E transmitted channels and the one or more cue codes (single channel mapped to two channels with appropriate ILD/ITD to each channel; col. 8, lines 1-14; see Fig. 6).

Regarding **claim 30,** Claim 30 is met since E=1(monophonic, col. 10, lines 8-12).

Regarding **claim 31,** Claim 31 is met since E>1(stereophonic, col. 10, lines 8-12).

Regarding **claim 32,** Claim 32 is met since each of the E transmitted channels is based on two or more of the C input channels (two channels, see Figs. 1, 4; using common transmit channel for all capture channels at a given endpoint; col. 5, lines 46-49; Fig. 5, col. 6, lines 1-10).

Regarding **claim 33,** Knappe further teaches wherein the one or more cue codes comprise one or more of inter-channel level difference data (ICLD), i.e., ILD (interaural level differences) and inter-channel time difference (ICTD) data i.e., ITD (interaural time delay; col. 7, lines 39-48; col. 8, lines 1-14).

Regarding **claim 34,** Knappe further teaches wherein the one or more cue codes comprise ICLD data, i.e., ILD (interaural level differences) and ICTD, i.e., ITD (interaural time delay, col. 7, lines 39-48; col. 8, lines 1-14) data.

Regarding **claim 35,** Knappe further teaches the invention of claim 27, wherein the downmixing comprises, for each of one or more different frequency bands, downmixing the two or more input channels in the frequency domain into one or more downmixed channels in the frequency domain (subband coding, col. 3, lines 54-64; col. 8, lines 44-63).

Regarding **claim 36,** Knappe further teaches the invention of claim 35, wherein the downmixing further comprises converting the one or more downmixed channels from the frequency domain into one or more of the transmitted channels in the time domain (reproduced by speakers 26L and 26R, Fig. 2, col. 4, lines 37-43).

Regarding **claim 37,** this claim merely reflects the means-plus-function to the method claim of claim 27 and is therefore rejected for the same reasons.

Regarding **claim 38** this claim merely reflects the apparatus to the method claim of claim 27 and is therefore rejected for the same reasons.

Regarding **claim 49,** this claim has similar limitations as Claim 27. Therefore it is interpreted and rejected under Knappe for the reasons set forth in the rejection of Claim 27. It is noted Knappe discloses encoded bitstream (see encoder 24, Fig. 1; col. 3, lines 54-64; col. 4, lines 52-59).

Regarding **claim 52,** this claim has similar limitations as Claim 27. Therefore it is interpreted and rejected under Knappe for the reasons set forth in the rejection of Claim 27. It is noted Knappe discloses encoded bitstream (see encoder 24, Fig. 1; col. 3, lines 54-64; col. 4, lines 52-59).

Claim Rejections - 35 USC § 103

5. The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all obviousness rejections set forth in this Office action:

> (a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negatived by the manner in which the invention was made.

6. **Claims 1-6, 9, 12-19, 22, and 25-72** are rejected under 35 U.S.C. 103(a) as being unpatentable over Ten Kate U.S. Patent 5,878,080 in view of Shaffer et al. U.S. Patent 6,973,184 (hereinafter, "Shaffer").

Regarding **claim 27**, Ten Kate teaches a method for encoding C (left, right and center channels, Figs.1a, 2) input audio channels to generate E (composite signal, combination signal; col. 2, lines 47-56) transmitted audio channels, the method (see Figs. 1a, 2, and respective portions of the specification), comprising:

providing two or more of the C input channels (3 channels, col. 2, lines 28-38) in a frequency domain (subband coding, col. 6, lines 47-59);

generating data (data compression, encoder; Figs. 1a, 2; col. 3, lines 61-66) for each of one or more different frequency bands in the two or more input channels in the frequency domain (col. 4, lines 20-35); and

combining (combining unit) the C input channels to generate the E (composite signal, combination signal) transmitted channels, where $C > E \geq 1$ (e.g., composite signal, combination signal; col. 2, lines 47-56) such that an audio decoder is enabled to generate more than E different playback audio channels based on the E transmitted channels (i.e., decoder is capable of also retrieving the one original signal transmitted and can retrieve the other two original signal, col. 2, lines 56-65).

Ten Kate further discloses generating data stream for transmission signal (col. 4, lines 20-35). However, Ten Kate does not explicitly disclose downmixing the input channels; data stream including one or more cue codes for each of two or more different frequency bands such that the audio decoder based on only the E transmitted channels and one or more cue codes.

Shaffer discloses encoding/decoding systems and methods for packet voice conferencing including downmixing the input channels (i.e., combine sound filed signals

as a single signal, monophonic, col. 2, lines 18-23); subband coding (col. 3, lines 44-47); cue codes for each of two or more different frequency bands (col. 5, lines 49-60) a directional cue for each speaker on the other end of the conference (col. 1, line 60 - col. 2, line 7; Figs. 4, 8); and such that the audio decoder based on only the E transmitted channels (i.e., monophonic) and one or more cue codes (col. 5, lines 38-50).

It would have been obvious to one of ordinary skill in the art at the time the invention was made to have incorporated encoding/decoding systems of Shaffer teaching with 3-channel encoder/2-channel decoder of Ten Kate so that downmixing the input channels; generating one or more cue codes for each of two or more different frequency bands; and such that the audio decoder based on only the E transmitted channels and one or more cue codes as claimed for purpose of decreasing bandwidth requirement, as suggested by Shaffer in column 3, lines 46-47).

Regarding **claim 28,** Ten Kate in view of Shaffer teaches the invention of claim 27. Ten Kate, as modified, further comprising formatting the E transmitted channels (composite signal, combination signal; col. 2, lines 47-56) and the one or more cue codes into a transmission format such that:

the format enables a first audio decoder (mono decoder, col. 2, lines 39-40) having no knowledge of the existence of the one or more cue codes (in combination signal) to generate E playback audio channels based on the E transmitted channels and independent of the one or more cue codes (mono output, col. 4, lines 36-39); and

the format enables a second audio decoder (standard stereo decoder, col. 2, lines 50-56) having knowledge of the existence of the one or more cue codes (in combination signal) to generate more than E playback audio channels based on the E transmitted channels and the one or more cue codes (col. 2, lines 50-59; col. 5, lines 17-24).

Regarding **claim 29,** Ten Kate in view of Shaffer teaches the invention of claim 28. Ten Kate, as modified, further teaches wherein the format enables the second audio decoder (standard stereo decoder, col. 2, lines 50-56) to generate C playback audio channels (two channels) based on the E transmitted channels and the one or more cue codes (col. 2, lines 50-59; col. 5, lines 17-24).

Regarding **claim 30,** Claim 30 is met since E=1(see Ten Kate, mono output, col. 4, lines 36-39).

Regarding **claim 31,** Claim 31 is met since E>1(see Ten Kate, stereo output, col. 4, lines 36-39; C=3-channel audio input; col. 2, lines 28-32).

Regarding **claim 32,** Claim 32 is met since each of the E transmitted channels is based on two or more of the C input channels (see Ten Kate, C=3-channel audio input; col. 2, lines 28-32).

Regarding **claim 33,** Shaffer, as modified, further teaches wherein the one or more cue codes comprise one or more of inter-channel level difference data (ICLD), i.e., ILD (interaural level differences) and inter-channel time difference (ICTD) data i.e., ITD (interaural time delay; see Shaffer, col. 4, lines 38-44).

Regarding **claim 34,** Shaffer, as modified, further teaches wherein the one or more cue codes comprise ICLD data, i.e., ILD (interaural level differences) and ICTD, i.e., ITD (interaural time delay, see Shaffer, col. 4, lines 38-44) data.

Regarding **claim 35,** Shaffer, as modified, further teaches the invention of claim 27, wherein the downmixing comprises, for each of two or more different frequency bands, downmixing the two or more input channels in the frequency domain into one or more downmixed channels in the frequency domain (see Shaffer, combine sound filed signals as a single signal, monophonic, col. 2, lines 18-23; subband coding col. 3, lines 44-47;).

Regarding **claim 36,** Shaffer, as modified, further teaches the invention of claim 35, wherein the downmixing further comprises converting the one or more downmixed channels from the frequency domain into one or more of the transmitted channels in the time domain (to output speakers 26L, 26R, Fig. 2).

Regarding **claim 1,** Ten Kate in view of Shaffer teaches the invention of claim 27, wherein:

the one or more cue codes comprise a plurality of scene parameters, i.e., ILD (interaural level differences) and i.e., ITD (interaural time delay; see Shaffer, col. 4, lines 38-44).

the E transmitted channels comprise a combined audio signal (composite signal, see Ten Kate, col. 2, lines 47-56);

generating the one or more cue codes and downmixing the C input channels comprises converting the input audio signals into the combined audio signal and the plurality of auditory scene parameters (i.e., ILD interaural level differences and ITD, interaural time delay; see Shaffer, col. 4, lines 38-44); and

further comprising embedding the auditory scene parameters into the combined audio, signal to generate an embedded audio signal (TRM signal, Fig. 2; see Ten Kate col. 10, line 66 – col. 11, line 9), such that:

a first receiver that is aware of the existence of the embedded auditory scene parameters (standard stereo decoder, see Ten Kate, col. 2, lines 50-56) can extract the auditory scene parameters from the embedded audio signal and apply the extracted auditory scene parameter to synthesize an auditory scene (col. 2, lines 50-59; see Ten Kate, col. 5, lines 17-24), and

a second receiver (mono decoder, see Ten Kate, col. 2, lines 39-40) that is unaware of the existence of the embedded auditory scene parameters can process the embedded audio signal (TRM signal, see Ten Kate, Fig. 2; col. 10, line 66 – col. 11, line

9) to generate an output audio signal, where the embedded auditory scene parameters are transparent to the second receiver (see Ten Kate, mono output, col. 4, lines 36-39).

Regarding **claim 2,** Ten Kate in view of Shaffer teaches the invention of claim invention of claim 1. Shaffer, as modified, further discloses wherein the plurality of auditory scene parameters comprise two or more different sets of one or more auditory scene parameters (i.e., ILD interaural level differences and ITD, interaural time delay; see Shaffer, col. 4, lines 38-44), wherein each set of auditory scene parameters corresponds to a different frequency band (subband coding, see Ten Kate, col. 6, lines 47-59) in the combined audio signal (composite signal, combination signal; col. 2, lines 47-56) such that the first receiver (standard stereo decoder, see Ten Kate, col. 2, lines 50-56) synthesizes the auditory scene by (a) dividing an input audio signal into a plurality of different frequency bands (subband coding, see Ten Kate, col. 6, lines 47-59); and (b) applying the two or more different sets of one or more auditory scene parameters (ILD interaural level differences and ITD, interaural time delay; see Shaffer, col. 4, lines 38-44) to two or more of the different frequency bands (subband coding, see Ten Kate, col. 6, lines 47-55) in the input audio signal to generate two or more synthesized audio signals of the auditory scene (including S_l, S_r, see Figs. 1a, 2) wherein for each of the two or more different frequency bands, the corresponding set of one or more auditory scene parameters is applied to the input audio signal as if the input audio signal corresponded to a single audio source in the auditory scene (S_l, S_r, M_o are applied to L, R, C; see Figs. 1a, 2, 3, 4; col. 4, line 40 – col. 5, line 15).

Regarding **claim 3,** Ten Kate in view of Shaffer teaches the invention of claim invention of claim 2. Shaffer, as modified, further teaches wherein each set of one or more auditory scene parameters corresponds to a different audio source in the auditory scene , i.e., ILD (interaural level differences) and i.e., ITD interaural time delay; (see Shaffer, col. 4, lines 38-44; microphones 20L, 20R simultaneously capture the sound field produced at two spatially-separated locations when B1, B2, or B3 talk, col. 3, lines 32-37).

Regarding **claim 4,** Ten Kate in view of Shaffer teaches the invention of claim invention of claim 2. Shaffer, as modified, further teaches wherein, for at least one of the sets of one or more auditory scene parameters, at least one of the auditory scene parameters corresponds to a combination of two or more different audio sources in the auditory scene that takes into account relative dominance of the two or more different audio sources in the auditory scene (arrival angle of the strongest source, see Shaffer, col. 5, lines 40-56).

Regarding **claim 5,** the invention of claim invention of claim 2. Shaffer, as modified, further teaches wherein the two or more synthesized audio signals comprise left and right audio signals of a binaural signal corresponding to the auditory scene (col. 4, lines 23-32).

Regarding **claim 6,** Ten Kate in view of Shaffer teaches the invention of claim invention of claim 2. Ten Kate, as modified, further teaches wherein the two or more synthesized audio signal comprise three or more signals of a multi-channel audio signal corresponding to the auditory scene (Fig. 3; col. 5, lines 52-61).

Regarding **claim 9,** Shaffer, as modified, further teaches wherein the auditory scene parameters comprise one or more of an interaural level difference, an interaural time delay, and a head-related transfer function (ILD interaural level differences and ITD, interaural time delay; see Shaffer, col. 4, lines 38-44).

Regarding **claim 14,** Ten Kate in view of Shaffer teaches the invention of claim 1, further teaches comprising steps of:

receiving the embedded audio signal comprising the combined audio signal embedded with the plurality of auditory scene parameters (i.e., ILD interaural level differences and ITD, interaural time delay; see Shaffer, col. 4, lines 38-44), wherein a receiver that is unaware of the existence of the embedded auditory scene parameters (mono decoder, see Ten Kate, col. 2, lines 39-40) can process the embedded audio signal to generate an output audio signal (see Ten Kate, mono output, col. 4, lines 36-39), where the embedded auditory scene parameters are transparent to the receiver (TRM signal, see Ten Kate, Fig. 2; col. 10, line 66 – col. 11, line 9)

extracting the auditory scene parameters from the embedded audio signal (col. 2, lines 50-59; see Ten Kate); and

applying the extracted auditory scene parameters to the combined audio signal to synthesize an auditory scene (col. 2, lines 50-59; see Ten Kāte, col. 5, lines 17-24).

Regarding **claim 15,** this claim has similar limitations as Claim 2. Therefore it is rejected for the reasons set forth in the rejection of Claim 2.

Regarding **claim 16,** this claim has similar limitations as Claim 3. Therefore it is rejected for the reasons set forth in the rejection of Claim 3.

Regarding **claim 17,** this claim has similar limitations as Claim 4. Therefore it is rejected for the reasons set forth in the rejection of Claim 4.

Regarding **claim 18,** this claim has similar limitations as Claim 5. Therefore it is rejected for the reasons set forth in the rejection of Claim 5.

Regarding **claim 19,** this claim has similar limitations as Claim 6. Therefore it is rejected for the reasons set forth in the rejection of Claim 6.

Regarding **claim 22,** this claim has similar limitations as Claim 9. Therefore it is rejected for the reasons set forth in the rejection of Claim 9.

Regarding **claim 55,** Ten Kate in view of Shaffer teaches the invention of claim 27. Shaffer as modified further comprising:

upmixing for each of two or more different frequency bands (at playout splitter 66, Fig. 8; see Shaffer, col. 9, lines 61-63) one or more of the E transmitted channels in a frequency domain to generate two or more of M playback channels in the frequency domain (split the monophonic channel into two channels, col. 5, lines 41-45; col. 9, lines 61-63);

applying the one or more cue codes to each of the one or more different frequency bands in the two or more playback channels in the frequency domain to generate two or more modified channels (produce stereo; col. 1, line 60 - col. 2, line 7; Figs. 4, 8; col. 5, lines 38-60); and

converting the two or more modified channels from the frequency domain into a time domain (to output speakers 26L, 26R, Fig. 2).

Regarding **claim 56,** Ten Kate in view of Shaffer teaches the invention of claim 55. Shaffer, as modified, further comprising:

prior to upmixing, converting the one or more of the E transmitted channels from the time domain to the frequency domain (decoding of subband coding, see Shaffer, col. 3, lines 38-50).

Regarding **claim 57,** Claim 57 is met since E=1(see Ten Kate, mono output, col. 4, lines 36-39).

Regarding **claim 58,** Claim 58 is met since E>1(see Ten Kate, stereo output, col. 4, lines 36-39; M=3-channel audio input; col. 2, lines 28-32).

Regarding **claim 59,** Claim 59 is met since each of the E transmitted channels is based on two or more of the M input channels (see Ten Kate, M=3-channel audio input; col. 2, lines 28-32) and at least one code cue (i.e., ILD, interaural level differences and ITD, interaural time delay, see Shaffer, col. 4, lines 38-44) .

Regarding **claim 60,** Shaffer, as modified, further teaches wherein the one or more cue codes comprise one or more of inter-channel level difference data (ICLD), i.e., ILD (interaural level differences) and inter-channel time difference (ICTD) data i.e., ITD (interaural time delay; see Shaffer, col. 4, lines 38-44).

Regarding **claim 61,** Shaffer, as modified, further teaches wherein the one or more cue codes comprise ICLD data, i.e., ILD (interaural level differences) and ICTD, i.e., ITD (interaural time delay, see Shaffer, col. 4, lines 38-44) data.

Regarding **claim 62,** Ten Kate in view of Shaffer teaches the invention of claim 55. Shaffer, as modified, further teaches wherein the upmixing (at playout splitter 66, Fig. 8) comprises, for each of two or more different frequency bands (subband coding,

col. 3, lines Left), upmixing at least two of the E transmitted channels in the frequency domain (at playout splitter 66, Fig. 8; see Shaffer, col. 9, lines 61-63).

Regarding **claim 37,** this claim merely reflects the means-plus-function to the method claim of claim 27 and is therefore rejected for the same reasons.

Regarding **claim 12,** this claim merely reflects the means-plus-function to the method claim of claim 1 and is therefore rejected for the same reasons.

Regarding **claim 13,** this claim merely reflects the apparatus to the method claim of claim 1 and is therefore rejected for the same reasons.

Regarding **claim 25,** this claim merely reflects the means-plus-function to the method claim of claim 14 and is therefore rejected for the same reasons.

Regarding **claim 26,** this claim merely reflects the apparatus to the method claim of claim 2 and is therefore rejected for the same reasons.

Regarding **claim 63,** this claim merely reflects the means-plus-function to the method claim of claim 55 and is therefore rejected for the same reasons.

Regarding **claims 38-47, and 64-71** these claims merely reflect the apparatus to the method claim of claims 27-36, and 55-62 and are therefore rejected for the same reasons.

Regarding **claim 48,** Claim 48 is met since Ten Kate in view of Shaffer as modified further teaches wherein:

the apparatus is a system selected from the group consisting of a digital video recorder, a digital audio recorder, a computer, a satellite transmitter, a cable transmitter, a terrestrial broadcast transmitter, and an entertainment system (laptop computer, see Shaffer, col. 2, lines 8-17); and

the system comprises the two or more filter banks (34L, 34R, Fig. 3; see Shaffer, col. 5, lines 61-67), the code estimator (parameter estimator 42, Fig. 3; see Shaffer, col. 6, lines 9-17), and the downmixer (i.e., combine sound filed signals as a single signal, monophonic, col. 2, lines 18-23)

Regarding **claim 72,** Claim 72 is met since Ten Kate in view of Shaffer as modified, further teaches wherein:

the apparatus is a system selected from the group consisting of a digital video recorder, a digital audio recorder, a computer, a satellite transmitter, a cable transmitter, a terrestrial broadcast transmitter, and an entertainment system (laptop computer, see Shaffer, col. 2, lines 8-17); and

the system comprises the two or more filter banks (34L, 34R, Fig. 3; see Shaffer, col. 5, lines 61-67), the code estimator (parameter estimator 42, Fig. 3; see Shaffer, col. 6, lines 9-17), and the upmixer (playout splitter 66, Fig. 8; see Shaffer, col. 9, lines 61-63).

Regarding **claim 49,** this claim has similar limitations as Claim 27. Therefore it is interpreted and rejected under Ten Kate in view of Shaffer for the reasons set forth in the rejection of Claim 27. It is noted Shaffer, as modified, discloses encoded bitstream (see Shaffer, col. 5, line 61 – col. 6, line 8)

Regarding **claim 50,** this claim has similar limitations as Claim 28. Therefore it is interpreted and rejected under Ten Kate in view of Shaffer for the reasons set forth in the rejection of Claim 28.

Regarding **claim 51,** this claim has similar limitations as Claim 29. Therefore it is interpreted and rejected under Ten Kate in view of Shaffer for the reasons set forth in the rejection of Claim 29.

Regarding **claim 52,** this claim has similar limitations as Claim 27. Therefore it is interpreted and rejected under Ten Kate in view of Shaffer for the reasons set forth in the rejection of Claim 27. It is noted Shaffer, as modified, discloses encoded bitstream (see Shaffer, col. 5, line 61 – col. 6, line 8)

Regarding **claim 53,** this claim has similar limitations as Claim 28. Therefore it is interpreted and rejected under Ten Kate in view of Shaffer for the reasons set forth in the rejection of Claim 28.

Regarding **claim 54,** this claim has similar limitations as Claim 29. Therefore it is interpreted and rejected under Ten Kate in view of Shaffer for the reasons set forth in the rejection of Claim 29.

7. **Claims 7 and 20** are rejected under 35 U.S.C. 103(a) as being unpatentable over Ten Kate U.S. Patent 5,878,080 in view of Shaffer et al. U.S. Patent 6,973,184 (hereinafter, "Shaffer"), further in view of Knappe et al. U.S. Patent 6,850,496 (hereinafter, "Knappe"), cited by Applicants.

Regarding **claim 7,** Ten Kate in view of Shaffer teaches the invention of claim invention of claim 1. Ten Kate in view of Shaffer, as modified, further teaches wherein the combined audio signal corresponds to a combination of two or more different source signals (composite signal from three original signals, see Ten Kate, col. 2, lines 33-56), wherein the two or more different frequency bands are selected by comparing magnitudes of the two or more different source signals (ILD, compare signal strength, see Shaffer, col. 5, lines 49-56), wherein, for each of the two or more different frequency

bands, one of the source signals dominates the one or more other source signals (arrival angle of the strongest source, see Shaffer, col. 5, lines 40-56).

Ten Kate in view of Shaffer does not explicitly disclose source signals are mono source signals.

Knappe discloses voice conferencing (col. 1, lines 7-8) in which source signals are mono source signals (each endpoint is considered a mono source signal since at a given endpoint a common transmit channel for all capture channels is preferred; see Fig. 4, col. 5, lines 46-49).

It would have been obvious to one of ordinary skill in the art at the time the invention was made to have incorporated the mono source signals taught by Knappe with method for encoding C input audio channels of Ten Kate in view of Shaffer such that the source signals are mono source signals as claimed for purpose of creating a perception of spatial separation between conference participants, as suggested by Knappe in column 2, lines 40-42.

Regarding **claim 20,** this claim has similar limitations as Claim 7. Therefore it is rejected for the reasons set forth in the rejection of Claim 7.

8. **Claims 8, 21 and 73-77** are rejected under 35 U.S.C. 103(a) as being unpatentable over Ten Kate U.S. Patent 5,878,080 in view of Shaffer et al. U.S. Patent 6,973,184 (hereinafter, "Shaffer"), further in view of Davis et al. U.S. Patent 6,850,496 (hereinafter, "Davis").

Regarding **claim 8,** Ten Kate in view of Shaffer teaches the invention of claim invention of claim 1. Ten Kate, as modified, further teaches, wherein the combined audio signal corresponds to a combination of left and right audio signals of a binaural signal (composite signal, see Ten Kate, col. 2, lines 47-56; col. 3, lines 61-66); Shaffer as modified further teaches wherein each different set of one or more auditory scene parameters (ILD interaural level differences and ITD, interaural time delay; see Shaffer, col. 4, lines 23-44) is generated by comparing (i.e., differences, time delay) the left and right audio signals in subband coding (see Shaffer, col. 3, lines 43-47).

However, Ten Kate in view of Shaffer does not explicitly disclose by comparing the left and right a corresponding frequency bands in subband coding.

Davis discloses an encoder generates a plurality of channel subband signals from a plurality of input signals in a plurality of frequency subbands, generates a spatial-characteristic signal representing spatial characteristics of a soundfield in response to respective channel subband signals in a frequency subband (see Fig. 1; col. 4, lines 18-33; col. 5, lines 60-67; col. 6, lines 8-33).

It would have been obvious to one of ordinary skill in the art at the time the invention was made to have incorporated the encoder taught by Davis with method for encoding C input audio channels of Ten Kate in view of Shaffer such that each of different set of one or more auditory scene parameters is generated as claimed for purpose of reducing the information requirements of signals that convey the audio information, as suggested by Davis in column 1, lines 11-12.

Regarding **claim 21,** this claim has similar limitations as Claim 8. Therefore it is rejected for the reasons set forth in the rejection of Claim 8.

Regarding **claim 73,** Ten Kate in view of Shaffer teaches the invention of claim 27. Shaffer, as modified, further, wherein the method comprises generating, in the frequency domain, ICTD data as one of the one or more cue codes (interaural time delay, for left channel, right channel, see Shaffer, col. 4, lines 11-67) for at least two of the two or more different frequency bands (subband coding,col. 3, lines 44-47).

However, Ten Kate in view of Shaffer does not explicitly disclose wherein each of the at least two different frequency bands has different ICDT data.

Davis discloses an encoder generates a plurality of channel subband signals from a plurality of input signals in a plurality of frequency subbands, spatial coder (40, Fig. 1) generates a spatial-characteristic signal representing spatial characteristics of a soundfield in response to respective channel subband signals in a frequency subband (see Fig. 1; col. 4, lines 8-33; col. 5, lines 60-67; col. 6, lines 8-33).

It would have been obvious to one of ordinary skill in the art at the time the invention was made to have incorporated the encoder taught by Davis with method for encoding C input audio channels of Ten Kate in view of Shaffer such that each of the at least two different frequency bands has different ICDT data as claimed for purpose of reducing the information requirements of signals that convey the audio information, as suggested by Davis in column 1, lines 11-12.

Regarding **claim 74,** this claim has similar limitations as Claim 73. Therefore it is interpreted and rejected for the same reasons.

Regarding **claim 75,** this claim has similar limitations as Claim 73. Therefore it is interpreted and rejected for the same reasons.

Regarding **claim 76,** this claim has similar limitations as Claim 73. Therefore it is interpreted and rejected for the same reasons.

Regarding **claim 77,** this claim has similar limitations as Claim 73. Therefore it is interpreted and rejected for the same reasons.

9. **Claims 10 and 23** are rejected under 35 U.S.C. 103(a) as being unpatentable over Ten Kate U.S. Patent 5,878,080 in view of Shaffer et al. U.S. Patent 6,973,184 (hereinafter, "Shaffer"), further in view of Wu et al. U.S. Patent 6,614,936 (hereinafter, "Wu").

Regarding **claim 10,** Shaffer, as modified, further teaches comprises the step of applying a layered coding technique (col. 3, lines 63-67) in which error protection (col. 1, lines 21-24) is provided to the combined audio signal than to the auditory scene parameters when generating the embedded audio signal (voice packet 50, Fig. 4);

encoder (24) and decoder (30) work as pair such have been obvious that errors due to transmission over a lossy channel will tend to affect the auditory scene parameters before affecting the combined audio signal to improve the probability of the first receiver to process at least the combined audio signal (Fig. 7, col. 8, lines 43-50).

Ten Kate in view of Shaffer does not explicitly disclose such coding technique having a strong error protection.

Wu discloses a layered coding technique in which providing a strong error protection (see col. 2, lines 26-32).

It would have been obvious to one of ordinary skill in the art at the time the invention was made to have incorporated the layered coding technique taught by Wu with method for encoding C input audio channels of Ten Kate in view of Shaffer such that providing a strong error protection as claimed for purpose of getting a particular quality according to its preference and capability, as suggested by Davis in column 2, lines 38-39.

Regarding **claim 23,** this claim has similar limitations as Claim 10. Therefore it is rejected for the reasons set forth in the rejection of Claim 10.

10. **Claims 11 and 24** are rejected under 35 U.S.C. 103(a) as being unpatentable over Ten Kate U.S. Patent 5,878,080 in view of Shaffer et al. U.S. Patent 6,973,184 (hereinafter, "Shaffer"), and further in view of Jafarkhani et al. U.S. Patent 6,823,018 (hereinafter, "Jafarkhani").

Regarding **claim 11**, Ten Kate in view of Shaffer teaches the invention of claim 1. Ten Kate, as modified, further teaches channel coding the serial datastream (col. 4, lines 23-30), a 2-channel information stream (col. 13, lines 2-4) in which in which the auditory scene parameters and the combined audio signal are both divided into two or more streams, wherein each stream divided from the auditory scene parameters is embedded into a corresponding stream divided from the combined audio signal to form a stream of the embedded audio signal (TRM signal, see Ten Kate, Fig. 2; col. 10, line 66 – col. 11, line 9), such that the two or more streams of the embedded audio signal may be transmitted over two or more different channels to the first receiver (standard stereo decoder, see Ten Kate, col. 2, lines 50-56) such that the first receiver is able to synthesize the auditory scene using extracted auditory scene parameters (col. 5, lines 17-24) when errors result from transmission of one or more of the streams of the embedded audio signal over one or more lossy channels (MPEG-1, col. 6, lines 47-59; col. 12, lines 19-29).

However, Ten Kate in view of Shaffer does not explicitly disclose applying multi-descriptive coding technique and using relative coarse resolution.

Jafarkhani discloses a multiple description coding communication system in which coarse (resolution) quantizer being used (col. 2, lines 28-49; col. 5, lines 38-42).

It would have been obvious to one of ordinary skill in the art at the time the invention was made to have incorporated a multiple description coding using coarse resolution of Jafarkhani teaching with coding C input audio channels of Ten Kate,

Shaffer in combination so that to apply a multiple description coding as claimed for purpose of providing reliably encoding and decoding information, as suggested by Jafarkhani in column 3, lines 7-9.

Regarding **claim 24,** this claim has similar limitations as Claim 11. Therefore it is rejected for the reasons set forth in the rejection of Claim 11.

Response to Arguments

11. Applicant's arguments with respect to claims 1-77 have been considered but are moot in view of the new grounds of rejection.

Conclusion

Any inquiry concerning this communication or earlier communications from the examiner should be directed to Con P. Tran whose telephone number is (571) 272-7532. The examiner can normally be reached on M - F (8:30 AM - 5:00 PM).

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor Vivian C. Chin can be reached on (571) 272-7848. The fax phone number for the organization where this application or proceeding is assigned is 571-273-8300.

Information regarding the status of an application may be obtained from the Patent Application Information Retrieval (PAIR) system. Status information for published applications may be obtained from either Private PAIR or Public PAIR.

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cpt CPJ
April 27, 2007

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